[CODING] APPARATUS FOR COMPRESSING AND EXPANDING A DIGITAL INPUT SIGNAL

BACKGROUND OF THE INVENTION

Field of the Invention

This invention relates to [a coding] an apparatus [adapted for implementing the so-called] that applies block floating processing to [an input digital signal, and to transform a signal on the time base to a signal on the frequency base to divide it into signal components in a plurality of critical bands to carry out bit allocation of the signal components every respective critical bands, thus to encode them.] a digital input signal, orthogonally transforms the block floating processed signal on the time axis into plural spectral coefficients on the frequency axis, divides the spectral coefficients into plural critical bands, and carries out adaptive bit allocation to quantize the spectral coefficients in each critical band.

Description of the Prior Art

As one of technologies for [implementing bit compression to an audio signal, etc. to encode it, there is known so called a block floating technology to divide input data into data blocks every] compressing a digital audio signal, and similar analog signals, it is known to apply block floating processing in which the digital input signal is divided into blocks of a predetermined number of [words to carry out a floating processing every respective blocks. In accordance with this block floating technology, an approach is employed to seek or search for the maximum one (maximum absolute value) of absolute values of respective words in a block to carry out a floating processing by using this maximum absolute value as a common] words, and block floating processing is applied to each block. In known block floating processing, the

maximum one of the absolute values of the words in the block is sought, and is used as a common block floating coefficient for all the words in the [corresponding] block. Further, [there] it is also known [an] to use orthogonal transform coding to transform [(so called] orthogonally [transform)] a signal on the time [base to] axis into a signal on the frequency [base to encode it. As this orthogonal transform coding technology, there is such a technology to divide, e.g., audio PCM data into data blocks every] axis. The resulting spectral coefficients are then quantized. For example, it is known to divide, e.g., a PCM audio signal into blocks, each of a predetermined number of [words to carry out] words, and to apply a Discrete Cosine Transform (DCT) [processing every respective blocks. In addition, there is also known such a technology to divide a signal on the frequency base into signal components every so called critical bands to apply an] to each block. In addition, it is also known to divide the spectral coefficients resulting from an orthogonal transform into critical bands and to quantize the spectral coefficients by applying adaptive bit allocation [thereto every respective bands.] to each critical band. The number of bits allocated [is determined in dependency upon allowed noise levels every critical bands in which the so-called masking is taken] for quantizing the spectral coefficients in each critical band is determined depending on an allowable noise level in each critical band, which takes psychoacoustic masking into consideration.

[Meanwhile, there] There are many instances where the operational processing for [such] an orthogonal [transformation] transform is executed by using a FIR (Finite-duration Impulse-Response) filter of the multi-tap type. [At this time, such an] This type of operational processing includes [a] coefficient multiplication processing and/or [an operation for taking a sum total, etc., resulting in the possibility that the number of digits may be increased so that it] operations for calculating a sum total, etc. The number of bits generated by such processing results the likelihood of overflows. To prevent such [an overflow, it is required to allow] overflows, the number of bits generated by

the operation must be allowed for in advance [the number of digits for operation] by, e.g., processing using several orders of [digits] bits greater than the number of [digits of] bits in each word of the input [data.] signal. For such a [multi-digit] multi-bit operation, a high performance DSP (Digital Signal Processing unit) is required, and it takes much time as well. Accordingly, [a more simple] simplification of the orthogonal transform processing is [expected.] desirable.

In view of this, a technique has been proposed to [implement] apply the above-mentioned block floating processing to the digital input [data] signal prior to the orthogonal transform processing. The block floating processing [to carry out]achieves bit compression [thereof to thereby reduce in advance] of the input signal and reduces the number of [digits of data] bits subject to the orthogonal transform operation.

Further, a technique has been also proposed to adaptively vary the [length of a] size of the block [for] subject to the orthogonal transform processing [in dependency upon an input and the above-mentioned technology for adaptively changing the length of a block in dependency upon an input signal are combined to carry out coding, respective processing are independently required, resulting in the drawback that a quantity subject to processing is increased.] depending on a signal. Such a technique is employed [because] because, particularly [in the case of dividing in advance an] when the input signal is divided into [signal] components in several (e.g., about three) [bands to carry out] frequency ranges, and the orthogonal transform processing [thereof every respective bands, a method of varying the length of a block in the magnitude of changes in time or a pattern, etc. of signals in respective bands] is performed in each frequency range, varying the block length in response to the magnitude of temporal changes, or in response to a pattern, etc., in the frequency range signals permits a more efficient [coding than a coding method in which the length of a block]quantizing of the resulting spectral components than when the block length is fixed.

It is to be noted that [in the case where the above-mentioned technology for implementing a] when block floating is applied prior to the orthogonal transform processing, and the block length is adaptively changed depending on a signal, independent processing is applied, which results in the drawback that the amount of processing required is increased. [processing and the above-mentioned technology for adaptively changing the length of a block in dependency upon an input signal are combined to carry out coding, respective processing are independently required, resulting in the drawback that a quantity subject to processing is increased.]

[In actual terms, for] For example, as shown in Fig. 15, a relatively large block BL is divided in advance into several sub blocks (e.g., the four [blocks), thus to prepare small]sub blocks BLS1, BLS2, BLS3 and [BLS4.] BLS4). As indicated by step S31 of Fig. 16, the respective energies of [these small] the sub blocks BLS1, BLS2, BLS3 and BLS4 are calculated [for determination of a] in the process of determining the size of [a]the variable length [block (block length) to determine, at] block. At the next step S32, [a] the block size [in dependency upon] is determined in response to the energies of the respective sub blocks. Then, at step S33, the maximum absolute value [thus calculated. At the next step S34, the orthogonal transform processing such as DCT for this block is carried out.] within the block determined in the previous step [Then, at step S33, a maximum absolute value within a determined block] is calculated to implement [a] block floating processing [on the basis of] using the calculated maximum absolute value. At the next step S34, orthogonal transform processing, such as DCT, is applied to the block.

In such a processing procedure, calculation of [energies every] the energy of each respective [small blocks] sub block BLS1, BLS2, BLS3 and [BLS4for determination of block size (block length),] BLS4 for determining the block size, and calculation of the maximum absolute values [of respective] in the thus-determined blocks for applying the block floating processing are required. As a result, [a] the quantity subject to processing or the number of

steps in processing by [the]a so-called microprogram is increased.

[Meanwhile, with respect to allowed noise levels determined every respective critical bands in consideration of the above-mentioned masking, a method] When determining an allowable noise level for each critical band to take account of masking, it has been proposed to correct [such allowed noise levels by taking] the allowable noise level to take into consideration the [so-called] minimum audible level characteristic [in the hearing sense] of the human [being. In accordance with this method, an allowed] sense of hearing. In this, an allowable noise level already calculated [and] is compared with a minimum audible [level are compared with each other to assume a greater one as a new allowed] level, and the greater level is selected as the new allowable noise level.

The [allowed] allowable noise level in which [the]masking is taken into consideration is [considered to be the same level in the above-mentioned] assumed to be constant across each critical band. However, since [a measured value of the minimum audible limit is given by using a sine wave, particularly in the region where the critical bandwidth is broad as in a high frequency band, a value at a low frequency portion and a value at a high frequency portion within the same critical band differ from each other to much degree. For this reason, if an approach is employed to give] the minimum audible level is measured using a sine wave, the can be an appreciable change in the minimum audible level between the low frequency end and the high frequency end of each critical band. This is particularly so at high frequencies, where the critical bands are relatively broad. For this reason, using a single minimum audible [limit level every critical band, an error would become large,] level for each critical band causes appreciable errors, resulting in the possibility [that there may take place useless number of allocated bits at] of an excess number of bits being allocated for quantizing the spectral coefficients towards the high frequency [portion within] end of the critical band.

In addition, although it is [a] conceivable to [finely] divide the critical

band into small [bands] sub bands, and to give a minimum audible [limits every small divided bands,] level for each sub band, this is not preferable [in that a] because the quantity of information required to be transmitted is increased.

SUMMARY OF THE INVENTION

[With such actual circumstances in view, this]This invention has been proposed, and [its.] its object is to provide [a coding] an apparatus for [a digital signal wherein in such cases of implementing a block floating] compressing a digital input signal. Block floating is applied prior to the orthogonal transform processing and [adaptively changing] the length of [a block in dependency upon an input signal, the] the block subject to transform processing is changed depending on a signal. The apparatus is constructed so that [it can reduce a] the quantity subject to [processing.]
processing is reduced.

Another object of this invention is to provide [a coding apparatus for a digital signal wherein in the case of dividing an input signal on the frequency base into signal components in critical bands to respectively apply adaptive bit allocation thereto on the basis of allowed noise levels, the apparatus is constructed so that it can reduce an error of a minimum audible level within a critical band when it is employed as the allowed noise level.] an apparatus for compressing a digital input signal in which, when the input signal is divided in frequency into spectral coefficients in critical bands, and adaptive bit allocation is applied thereto on the basis of allowable noise levels, errors in the minimum audible level are reduced in those critical bands in which the minimum audible level is selected as the allowable noise level.

[To achieve the above-mentioned objects, according to a first aspect of this invention, there is provided a coding apparatus for a digital signal comprising: a band division filter for dividing an input digital signal into signal components in a plurality of frequency bands; a block floating circuit for implementing, every block, floating processing to an output signal of the band division filter; a plurality of orthogonal transform circuits for orthogonally

transforming respective output signals on the time base of the block floating circuit to signals on the frequency base; and an adaptive bit allocation encoder for dividing output signals of the orthogonal transform circuits into signal components in critical bands to adaptively allocate bit numbers thereto on the basis of allowed noise levels every respective critical bands, wherein the length in the time base direction of the block is caused to be variable, and the length in the time base direction of the block and floating coefficients at the time of the floating processing are determined on the basis of the same index.]

[The above-mentioned index may be given by a logical sum of absolute values of respective words.]

[Further, according to a second aspect of this invention, a coding apparatus for a digital signal featured by the first aspect of the invention may further comprise allowed noise level calculation means for calculating allowed noise levels obtained within the critical band every critical band; and comparison means for comparing the allowed noise level with a minimum audible level to raise or set a flag when the minimum audible level is higher than the allowed noise level, wherein, in the critical band where the comparison raises the flag, the level of the minimum audible curve is selected as the allowed noise level.]

[In the coding apparatus of the second aspect, the allowed noise level calculation means may be constructed to calculate an allowed noise level from an energy every critical band and a minimum audible curve, etc., and to further calculate an allowed noise level on the basis of an error between an output information quantity and a bit rate target value of the final coded data. Such an approach may be employed to increase or decrease allocated bits of respective unit blocks by using an output of the allowed noise level calculation means.]

[Furthermore, the feature of the first aspect may be combined with the feature of the second aspect.] [Namely, in the above-mentioned configuration of the coding apparatus of the second aspect, the length in a time direction of the block may be caused to be variable, and the length in the time

direction of the block and floating coefficients at the time of the floating processing may be determined on the same index.]

[In addition, the above-mentioned orthogonal transform circuit may be constructed as a Discrete Cosine Transform (DCT) circuit.]

[As taught by the above-mentioned aspects of this invention, the coding apparatus for digital signal of the invention is adapted to carry out a block floating processing of an input digital signal by a variable length block thereafter to implement an orthogonal transform processing thereto. In this coding apparatus, by determining the length of a variable length block and a floating coefficient of the block floating on the basis of the same index, thereby making it possible to reduce a quantity subject to quantization or the number of steps of a program.]

Accordingly, a first aspect of the invention provides an apparatus for compressing a digital input signal. The apparatus comprises an index generating circuit that generates an index in response to the digital input signal. Also included in the apparatus are a block length decision circuit, which determines a division of the digital input signal into blocks in response to the index, and a block floating processing circuit, which applies block floating processing to the blocks of the digital input signal in response to the index. The circuit further includes an orthogonal transform circuit that orthogonally transforms each block floating processed block of the digital input signal to produce plural spectral coefficients. Finally, the circuit comprises an adaptive bit allocation circuit that divides the plural spectral coefficients into bands, and adaptively allocates a number of quantizing bits to quantize the spectral coefficients in each of the bands.

A variation of the first aspect of the invention provides an apparatus for compressing a digital input signal. The apparatus comprises a band division filter that divides the digital input signal into a frequency range signal in each of plural frequency ranges. Also included in the apparatus are a block length decision circuit, which determines a division of each frequency range signal in

time into blocks in response to an index, and a block floating processing circuit, which applies block floating processing to each frequency range signal in response to the index. The circuit also includes an orthogonal transform circuit that orthogonally transforms each block floating processed frequency range signal to produce plural spectral coefficients. The orthogonal transform circuit transforms each frequency range signal in blocks determined by the block length decision means. Finally, the apparatus comprises an adaptive bit allocation circuit that divides the plural spectral coefficients into bands, and adaptively allocates numbers of quantizing bits for quantizing the spectral coefficients in response to an allowable noise level in each of the bands.

A second aspect of the invention provides an apparatus for compressing a digital input signal. The apparatus comprises a circuit that derives plural spectral coefficients from the digital input signal, and an adaptive bit allocation circuit that divides the spectral coefficients by frequency into bands, and adaptively allocates a number of quantizing bits for quantizing the spectral coefficients in each band in response to an allowed noise level for each of the bands. The adaptive bit allocation circuit includes an allowable noise level calculation circuit that calculates an allowed noise level for each band, a comparator that compares the allowable noise level with a minimum audible level in each band, and a selector that selects the minimum audible level as the allowable noise level for each band in which the comparator determines that the minimum audible level is higher than the allowable noise level.

A variation on the second aspect of the invention provides an apparatus for compressing a digital input signal. The apparatus comprises a band division filter that divides the digital input signal into a frequency range signal in each of plural frequency ranges. Also included in the apparatus are a block floating processing circuit that applies block floating processing to each frequency range signal divided in time into blocks, and an orthogonal transform circuit that orthogonally transforming each block of each frequency range signal to provide plural spectral coefficients. Finally, the apparatus includes an adaptive bit

allocation circuit that divides the spectral coefficients into bands, and adaptively allocates a number of quantizing bits for quantizing the spectral coefficients in each band in response to an allowable noise level in each band. The adaptive bit allocation circuit includes an allowable noise level calculation circuit that calculates the allowable noise level for each band, and a comparator for comparing the allowable noise level with a minimum audible level in each band, and that sets a flag for each band in which the minimum audible level is higher than the allowable noise level. Finally, the adaptive bit allocation circuit includes a selector that selects the minimum audible level as the allowed noise level in each band in which the flag is set.

[Further, in accordance with the coding apparatus for digital signal featured above, when an allowed noise level every critical bands] When the allowable noise level in each critical band is determined by the minimum audible level, bit allocation is carried out [by allowed noise levels every small]according to the allowable noise level in plural sub bands obtained by further dividing the critical band [to only transmit]in frequency. When this is done, a flag indicating [this, thus to avoid] that the minimum audible level has been adopted as the allowable noise level for the band only needs to be transmitted. This avoids the necessity of [sending allowed noise levels every small bands.] transmitting allowable noise level information for each sub band. Accordingly, accurate [allowed] allowable noise levels can be provided without increasing the quantity of auxiliary information [quantity. This leads to the fact that]transmitted. This provides an improvement in signal quantity [can be improved] without degrading [bit] the signal compression efficiency. In addition, even if [an] the absolute value of the minimum audible [limit] level is altered later, compatibility can be maintained.

A third aspect of the invention provides a method for compressing a digital input signal. In the method, an index is generated in response to the digital input signal, a division of the digital input signal into blocks is determined in response to the index, and block floating processing is applied to

the blocks of the digital input signal in response to the index. Each block floating processed block of the digital input signal is orthogonally transformed to produce plural spectral coefficients, the spectral coefficients are divided into bands, and numbers of quantizing bits are adaptively allocated to quantize the spectral coefficients in each band.

A fourth aspect of the invention provides a method for compressing a digital input signal. In the method, plural spectral coefficients are derived from the digital input signal, the spectral coefficients are divided by frequency into bands, and a number of quantizing bits is allocated for quantizing the spectral coefficients in each band in response to an allowed noise level for each band. In the step of adaptively allocating a number of quantizing bits, an allowable noise level is calculated for each band, the allowable noise level is compared with a minimum audible level in each band, and the minimum audible level is selected as the allowable noise level in each band in which the minimum audible level is higher than the allowable noise level.

A fifth aspect of the invention provides an apparatus for expanding a compressed digital signal. The compressed digital signal includes plural quantized spectral coefficients and auxiliary information. The apparatus comprises adaptive bit allocation decoding circuit that operates in response to the auxiliary information and inversely quantizes the quantized spectral coefficients to provide plural spectral coefficients. The circuit also includes a block floating circuit that applies block floating to the spectral coefficients. Also included in the apparatus is an inverse orthogonal transform circuit means that inversely orthogonally transforms the block floating processed spectral coefficients to provide plural frequency range signals. Finally, the apparatus includes an inverse filter circuit that synthesizes the frequency range signals to provide an output signal.

A sixth aspect of the invention provides a method for expanding a compressed digital signal to provide a digital output signal. The compressed digital signal includes plural quantized spectral coefficients divided by

frequency into bands. At least one of the bands is a divided band in which the spectral coefficients in the band are further divided by frequency into sub bands. The compressed digital signal additionally includes an allowed noise level for each band, and, for each divided band, a flag signal. The quantized spectral coefficients in each band and sub band are quantized using an adaptively- allocated number of quantizing bits. In the method, in each divided band, the allowed noise level of the band is set as the allowed noise level for the band when the flag signal for the band is in a first state. Also, in each divided band, the allowed noise level of the band is set as the allowed noise level for one of the sub bands constituting the band when the flag signal for the band is in a second state. Finally, in each divided band, an allowed noise level for each of the other sub bands constituting the band is calculated from the allowed noise level of the band. The allowable noise level for each band and sub band is then used to inversely quantize the respective quantized spectral coefficients in each band and sub band, and the digital output signal is derived from the resulting spectral coefficients.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a circuit diagram showing, in a block form, the outline of the configuration of [a coding] an apparatus for compressing a digital input signal according to an embodiment of this invention.

Fig. 2 is a view showing an actual example of [divided bands and formation of blocks in the time base direction in the respective bands] how the input signal is divided into frequency ranges and how the input signal is divided in time into blocks in each frequency range in the embodiment.

Fig. 3 is a [flowchart] flow chart for explaining the essential part of [an encoding operation] the process by which the allowable noise level is set in the embodiment.

Fig. 4 is a view showing a critical band used for explaining [an encoding operation] how the allowable noise level is set in the embodiment.

Fig. 5 is a [flowchart] flow chart for explaining the essential part of [a] the decoding operation in the embodiment.

Fig. 6 is a view showing a critical band used for explaining [a] the decoding operation in the embodiment.

Fig. 7 is a view showing [the] an example [where a] in which the block size in [the time base direction in respective bands] one frequency range is switched between two sizes in the apparatus of Fig. 1.

Fig. 8 is a view showing [the] an example [where a] in which the block size in [the time base direction in respective bands] one frequency range is switched between three sizes in the apparatus of Fig. 1.

Fig. 9 is a [flowchart] flow chart for explaining the block floating operation of the embodiment.

Fig. 10 is a circuit diagram showing, in a block form, an actual example of [allowed] allowable noise calculation circuit 20 of the apparatus shown in Fig. 1.

Fig. 11 is a view showing a bark spectrum.

Fig. 12 is a view showing a masking spectrum.

Fig. 13 is a view in which a minimum audible level curve and a masking spectrum are synthesized.

Fig. 14 is a block diagram showing an actual example of a decoder to which the embodiment of this invention can be applied.

Fig. 15 is a view showing an example of the length of a block by the processing procedure in the prior art.

Fig. 16 is a [flowchart] flow chart showing an example of the procedure of a conventional block floating processing.

DESCRIPTION OF THE PREFERRED EMBODIMENT

A preferred embodiment of this invention will now be described with reference to the attached drawings.

[A coding apparatus for a digital signal to which this] This invention can

be applied [is directed to coding apparatuses for a digital signal, which are adapted for carrying out an efficient coding of an input digital signal such as an audio PCM signal, etc. by using respective technologies of subband division] to an apparatus for compressing a digital input signal, such as a PCM audio signal, etc., using subband coding (SBC), adaptive transform coding (ATC), and adaptive bit allocation (APC-AC). In [actual terms, in] the apparatus of the embodiment shown in Fig. 1, [a technique is employed to divide an input digital signal into signal components in a plurality of frequency bands, and to make a setting such that the bandwidths become broader according as the frequency shifts to a higher frequency band side to carry out orthogonal transform processing every respective frequency bands to apply adaptive bit allocation to spectrum data on the frequency base thus obtained every so called critical band in which the hearing sense] the digital input signal is divided into frequency range signals in plural frequency ranges. The bandwidths of the frequency ranges increase with increasing frequency. Orthogonal transform processing is applied to each frequency range signal to provide plural spectral coefficients. Adaptive bit allocation is used to quantize the spectral coefficients divided into critical bands in which the masking characteristic of the human [being]sense of hearing is taken into [consideration which will be described later, thus to encode them.] consideration. In addition, in the embodiment of this invention, [a technique is also employed to] the block size [(block length) in dependency upon an input] is adaptively [vary a] varied in response to a [block size (block length) in dependency upon an input] signal prior to the orthogonal transform processing, and [to carry out a] block floating processing is applied to every block.

[Namely, in FIG.] In Fig. 1, input terminal 10 is supplied with [an] a PCM audio [PCM] signal in the frequency range of, e.g., 0 Hz to 20 [KHz.] kHz. This input signal is divided into a signal in the frequency [band of 0] range of 0 Hz to 10 [KHz]kHz and a frequency range signal in the frequency [band] range of 10 [K] to 20 [KHz] kHz by using a band division filter 11,

e.g., [the so-called QMF filter, etc., and the] a Quadrature Mirror Filter (QMF filter), etc. The signal in the frequency [band of 0] range of 0 Hz to 10 [KHz] kHz is further divided into a [signal in the frequency band of 0 to 5 KHz and a signal in the frequency band of 5 k to 10 KHz similarly] frequency range signal in the frequency range of 0 Hz to 5 kHz and a frequency range signal in the frequency range of 5 to 10 kHz by using a band division filter 12, e.g., [the so-called] a QMF filter, etc. The signal in the frequency [band] range of 10 [K] to 20 [KHz] kHz from the band division filter 11 is sent to [a] the Discrete Cosine Transform (DCT) circuit 13 serving as an orthogonal transform circuit, the signal in the frequency [band of 5 K] range of 5 to 10 [KHz] kHz from the band division filter 12 is sent to [a] the DCT circuit 14, and the signal in the frequency [band of 0 to 5 KHz] range of 0 Hz to 5 kHz from the band division filter 12 is sent to [a] the DCT circuit 15. Thus, these signals are subjected to DCT processing, respectively.

[Meanwhile, in] In the embodiment of this invention, in order to reduce [a] the quantity of [operation]operations in the orthogonal transform processing, [a technique is employed to implement] block floating processing [to an input data on the time base which has not yet been subjected to the orthogonal transform processing to carry out bit compression thereof to release the above-mentioned block floating after that data has been subjected] is applied to the frequency range signals prior to the orthogonal transform processing. This provides data compression. The block floating is released after the block floating processed signals have been orthogonally transformed.

[Namely, in Fig. 1, data on the time base of respective bands] In Fig. 1, the frequency range signals obtained from the band division filters 11 and 12 are delivered to a block floating processing circuit 16, [at] in which [a] block floating processing is carried out [with] using the respective blocks BL as shown in Fig. [15 being as a unit. At respective orthogonal] 15. In the transform circuits (DCT, i.e., Discrete Cosine [Transform] Transform, circuits are shown in the example of Fig. 1) 13, 14 and 15, [as operation for the]

orthogonal transform processing is [implemented] applied to the [data] signals which have undergone such block floating processing. [Thereafter release the above-mentioned] Thereafter, the block floating is released by [a] the block floating release circuit 17. In releasing the block floating, block floating information from the block floating processing circuit 16 is used. [Also in the case of determining floating coefficients in such a block floating processing, an approach may be adopted to take a logical sum of absolute values of respective words in a block as previously described.] Block floating coefficients may be determined in the block floating processing by taking the logical sum of the absolute values of the words in each block.

[An actual example of a standard input signal with respective to blocks every respective frequency bands delivered to the DCT circuits 13, 14 and 15 is shown in FIG. 2. In the actual example of FIG. 2, a scheme is employed such that according as the frequency shifts to a higher frequency band side, the frequency bandwidth is caused to be broad and the time resolution is caused to be high (the block length is caused to be short). Namely, for a signal in the frequency band of 0 to 5 KHz on a lower frequency band side, one block BL1 is caused to have, e.g., 1024 samples. For a signal in the medium frequency band of 5 K to 10 KHz, that signal is divided into signal components in blocks BLM1 and BLM2 each having a length TBL/2 one half of a length TBL of the block BL1 on the low frequency band side. For a signal in the frequency band of 10 K to 20 KHz on a higher frequency band side, that signal is divided into signal components in blocks BLH1, BLH2, BLH3 and BLH4 each having a length TBL/4 one fourth of that of the block BLL on the lower frequency band side. It is to be noted that in the case where the frequency band of 0 to 22 KHz is taken into consideration for an input signal, the low frequency band is in a range from 0 to 5.5 KHz, the medium frequency band is in a range from 5.5 K to 11 KHz, and the high frequency band is in a range from 11 K to 22 KHz.]

Fig. 2 shows an actual example of how a frame of the digital input

signal is divided into blocks in each frequency range prior to delivery to the DCT circuits 13, 14 and 15. In the actual example of Fig. 2, the bandwidth of the frequency ranges increases and the time resolution increases (i.e., the block size is reduced) as the frequency increases. For the frequency range signal in the low frequency range of 0 Hz to 5 kHz, one block BLL is chosen to have, e.g., 1024 samples. For the frequency range signal in the middle frequency range of 5 to 10 kHz, the frame is divided into two blocks BLM1 and BLM2, each having a length TBL/2, one half of the length TBL of the block BLL in the low frequency range. For the frequency range signal in the high frequency range of 10 to 20 kHz, the signal is divided into four blocks BLH1, BLH2, BLH3 and BLH4, each having a length TBL/4, one fourth of the length TBL of the block BLL of the low frequency range. It is to be noted that, in the case where the input signal has a frequency range of 0 Hz to 22 kHz, the low frequency range extends from 0 Hz to 5.5 kHz, the middle frequency range extends from 5.5 to 11 kHz, and the high frequency range extends from 11 to 22 kHz.

[It is to be noted that, in] In the embodiment of this invention, as will be described later, the block size [(block length)] is caused to [be variable in dependency upon an input] vary in response to a signal, and determination of the block size is carried out [on the basis of a] in response to the maximum absolute value used also for determining the block floating coefficients of the block floating.

Turning back to [FIG. 1, spectrum data or DCT coefficient data on the frequency base,] Fig. 1, the spectral coefficients obtained as the result of the DCT processing [at] in the respective DCT circuits 13, 14 and 15, are [subjected to releasing of the] subject to block floating release processing [at the] in the block floating release circuit 17, and are then [combined every the so-called critical band. The data thus obtained are further sent to an adaptive bit allocation encoder 18. This critical band is a frequency band divided in consideration of the hearing sense characteristic of the human being, and is a

band that a narrow band noise having the same intensity as that of a pure sound in the vicinity of a frequency thereof has when the pure sound is masked by that noise. This critical band is such that according as the frequency shifts to a higher frequency band side, the bandwidth becomes broader, and the entire frequency band of 0 to 20 KHz is] divided by frequency into critical bands. The spectral coefficients are sent to the adaptive bit allocation circuit 18.

A critical band is division of the frequency range that takes into account characteristics of the human sense of hearing. A critical band is the band of noise that can be masked by a pure signal that has the same intensity as the noise and has a frequency in the middle of the critical band. The bandwidth of successive critical bands increases with increasing frequency. The audio frequency range of 0 Hz to 20 kHz is normally divided into, e.g., 25 critical bands.

[An allowed] The allowable noise calculation circuit 20 calculates [allowed noise quantities every respective critical bands in which the so-called masking effect, etc. is taken into consideration on the basis of spectrum data divided every critical band to calculate allocated bit numbers every respective critical bands on the basis of the allowed noise quantities and energies or peak values, etc. every respective critical bands. In dependency upon bit numbers allocated every respective critical bands] an allowable noise level for each critical band, taking into account the masking effect. The spectral coefficients are divided into plural critical bands to calculate the number of bits to be allocated to quantize the spectral coefficients in each critical band. Quantizing bits are allocated on the basis of the allowable noise level and the energy or peak value, etc. in each critical band. In response to the bit numbers allocated to each critical band by the adaptive bit allocation [encoder 18, respective spectrum data (or DCT coefficient data) are requantized. Data thus coded] circuit 18, the spectral coefficients are quantized. The quantized spectral coefficients are taken out through the output terminal 19.

[Here, the above-mentioned allowed] The allowable noise calculation

circuit 20 is supplied with a minimum audible [levels every respective bands] level for each critical band from a minimum audible level curve generator 32. Each minimum audible level is compared with [an allowed] the allowable noise [level] level, in which the [above-mentioned] masking effect is taken into [consideration at a] consideration, in the comparator 35. As a result, when the minimum audible level is higher than the [allowed noise level, this minimum audible level is regarded as an allowed noise level. At this time, an approach is employed to divide the critical band into smaller regions by taking into consideration an error of the minimum audible level particularly in bands where the critical bandwidth is broad to allow minimum audible levels every these small divided bands to be respective allowed noise levels, thus to carry out bit allocation every respective small divided bands.] allowable noise level, the minimum audible level is selected as the allowable noise level.

According to the invention, some critical bands, especially the higher frequency critical bands, are divided into sub bands to take into consideration the error in the minimum audible level that occurs particularly in critical bands having a wide bandwidth. Dividing critical bands allows a minimum audible level for each sub band to be used for the respective allowable noise level for each sub band. Bit allocation is then carried out for each sub band.

[The operation thereof] The operation of the division into sub bands will now be described with reference to Figs. 3 and 4.

Fig. 3 is a [flowchart] flow chart for explaining the operation, and Fig. 4 shows the example where one critical band [8] B is divided into [smaller regions] sub bands BB (four [regions] sub bands are shown in the example of Fig. 4).

[Initially, a] In the step S1 of Fig. 3, [there is carried out discrimination as to] it is determined whether or not the level of the minimum audible level curve RC of the [small]sub band [BB1 on] BB1, the lowest frequency [side] sub band of the four [small] sub bands BB1 to BB4 of [one] the critical band [B] B, is higher than the [level of a masking spectrum which is a present

allowed] masking level, which is the present allowable noise determined in consideration of [the] masking (RC > MS). [When the discriminated] If the result [at] of this step [St] S1 is YES (the level of the minimum audible level curve RC is higher than the masking [spectrum] level [of the masking spectrum] MS), the operation proceeds to step S2 [to consider the allowed noise as a] where the minimum audible [curve RC to raise or set a] level is selected as the allowable noise level. The flag FRC is set at the next step S3 [(FFC=1). Subsequently, the] (FRC=1). The operation proceeds to step S4 [to conduct an] where adaptive bit allocation [in dependency upon] is carried out using the level of the minimum audible [curve RC which is an allowed noise to carry out coding.] [On the contrary, when the discriminated result at] level curve RC as the allowable noise level. Conversely, when the result of the step S1 is NO, the operation proceeds to step S5 [to consider the allowed noise as a masking spectrum to set the flag FRC to O at step S6 to proceed to the above-mentioned step S4.] where the masking level is selected as the allowable noise level. The flag FRC is cleared to 0 at step S6, and the process reverts to the step S4 where adaptive bit allocation is carried out.

[Here, when attention is drawn to one critical band B as shown in Fig. 4, the case where the minimum audible curve is RCa with respect to the masking spectrum MS as an allowed noise obtained at present corresponds to the case where the discriminated result at the step S1 is YES, and the case where the minimum audible curve is RCb or RCc corresponds to the case where the discriminated result at] Various possibilities for one critical band B are shown in Fig. 4. Where the minimum audible level curve is the curve RCa and the masking level is the level MS, the result of the step S1 is YES. Where the minimum audible level curve is the curve RCb or the curve RCc, the result of the step S1 is NO. When the minimum audible [curve is RCa, this minimum audible curve RCa becomes an allowed noise. Thus, bit allocation is carried out every small bands BB1 to BB4 in dependency upon allowed noise levels given every small bands BB1 to BB4. On the contrary, when the minimum

audible curve is RCb or RCc, the allowed noise becomes the masking spectrum MS. Thus, bit allocation is carried out in dependency upon a single allowed noise level in the critical band B.] level curve is the curve RCa, the minimum audible level curve RCa is selected as the allowable noise level, and bit allocation is carried out in each sub band BB1 to BB4 in response to the allowable noise level in each sub band BB1 to BB4. On the other hand, when the minimum audible level curve is the curve RCb or the curve RCc, the masking level MS is selected as the allowable noise level, and bit allocation is carried out in response to a single allowable noise level throughout the whole critical band B.

[Meanwhile, in the case of transmitting the allowed noise level as auxiliary information along with quantized main information, even when the minimum audible curve RCa is considered as an allowed noise, information transmitted is only a single allowed noise level in the critical band. This is because since the minimum audible curve is determined from the hearing sense characteristic of the human being, a minimum audible curve pattern or relative value data, etc. is caused to be stored in advance into a ROM, etc., thereby making it possible to easily determine the minimum audible level of other small bands BB2 to BB4 basis of] The allowable noise level for each critical band is transmitted from the compressor as auxiliary information, along with quantized spectral coefficients as main information. This is so, even when the minimum audible level curve RCa is selected as the allowable noise level. The auxiliary information transmitted is a single allowable noise level for each critical band. The minimum audible level curve is determined from the characteristics of the human sense of hearing. Thus, a minimum audible level curve pattern, or relative value data, etc., can be stored in advance into a ROM, etc. The minimum audible level of the other sub bands BB2 to BB4 can easily be determined from the data in the ROM in response to the minimum audible level of, e.g., the [small] lowest-frequency sub band BB1.

Fig. 5 is a [flowchart] flow chart for explaining [the] an essential part of

the [decoding] expansion processing [on a decoder side.] in a complementary expander. At step S11 of Fig. 5, [discrimination as to] it is determined whether or not the flag FRC is [1 is made. When the discriminated] 1. If the result is YES, i.e., [an allowed] the allowable noise level of the corresponding critical band is given by the minimum audible [curve, allowed noise levels every respective small bands BB1 to BB4 are] level curve, the allowable noise level for each sub band BB1 to BB4 is calculated at the next step S12. [Namely, even if]Even though only one [allowed] allowable noise level [with respect to a single] is transmitted for all of the critical band B, e.g., [an allowed] the allowable noise level NL1 of the [small band BB1 on the] lowest frequency [side is sent] sub band BB1, as shown in Fig. 6, [allowed] allowable noise levels NL2 to NL4 [every respective small] for the other sub bands BB2 to BB4 can be determined by calculation from the pattern of the minimum audible level curve RC by making use of a relative list, etc. of minimum audible level values stored in a ROM, etc. as described above.

[Further, when the discriminated] If the result at the step S11 is NO, i.e., [an allowed noise of the corresponding] the allowable noise level for the critical band is given by the masking [spectrum] level MS, the operation proceeds to step S13 [to set a fixed allowed noise in a single] where a fixed allowable noise level is set for the whole of the critical band [8. On the basis of the allowed noise levels determined at these respective steps S12 and S13, bit] B. Bit allocation decoding processing [is executed at the next step S14.] takes place at step S14 in response to the allowable noise level determined at the respective one of the steps S12 and S13.

[Meanwhile, the lengths of respective blocks (block sizes) in forming blocks in the time base direction every respective frequency bands divided by the above-mentioned division filters 11 and 12 to carry out floating processing thereafter to conduct orthogonal transform processing are adaptively switched in dependency upon an input signal.]

The method of determining the division of each frame of each frequency

range signal into the blocks in which the respective frequency range signals are orthogonally transformed, and the way in which the block sizes are adaptively changed in response to a signal, i.e., each respective frequency range signal, will now be described.

[Namely, explanation will be given in connection with the case where block size switching between a block BL having a large time width TBL and blocks BLR1 and 8LR2] First, the case where the division of the frame is switched between a block BL, having a block length TBL, and two blocks BLR1 and BLR2 each having a block length of [TBL/2] TBL/2, one half of [TBL as shown in] TBL, will be described with reference to Fig. 7. First, maximum absolute values (or logical sums) MXR1 and MXR2 in the respective sub blocks [with respect to]corresponding to the smaller blocks BLR1 and BLR2 are determined. Then, [comparison between] these maximum absolute values MXR1 and MXR2 [is made.] are compared. When the ratio therebetween is as indicated by the following equation [(1);] (1), the frame is divided into the [switching to the size of] smaller blocks BLR1 and [BLR2 is made.] BLR2.

MXR2/MXR1 20 . . . (1)

[switching to the size of smaller blocks BLR1 and BLR2 is made. When otherwise,] Otherwise, a block size equal to the size of the larger block BL is selected.

[Then, explanation will be given in connection with] Next, the case where [block size switching] the division of the frame is switched between a large block BL having a [time width] block length TBL, medium blocks BLR1 and BLR2 having a block length [TBL/2] TBL/2, one half [thereof,] of the block length TBL, and small blocks BLS1, BLS2, BLS3 and [BLS4] BLS4, each having a block length [TBL/4 one half thereof.] TBL/4, one fourth of the block length TBL, will be described with reference to Fig. 8. First, respective maximum absolute values (or logical sums) MXS1, MXS2, MXS3 and [XS4 in] MXS4 in the sub blocks [of] corresponding to the small blocks BLS1, BLS2,

BLS3 and BLS4 are determined. With respect to these four maximum absolute values MXS1, MXS2, MXS3 and MXS4, [when] if the following relationship [as] indicated by the following equation (2) [holds;] holds, the frame is divided into blocks equal to the small blocks BLS1, BLS2, BLS3 and BLS4, having a length of TBBL/4.

MXSn+1/MXSn 20 . . . (2)

where n is 1, 2 or [3,] 3.

[the block size of the small blocks BLS1, BLS2, BLS3 and BLS4 having a length of TBBL/4 is selected. In contrast, when] If the above equation (2) is not satisfied, the respective maximum absolute values (or logical sums) MXR1 and MXR2 in the sub blocks [of] corresponding to the medium blocks BLR1 and [BLR2. Whether] BLR2 are determined. Then it is determined whether or not the following equation (3) is [satisfied is discriminated.] satisfied.

MXR2/MXR1 10 . . . (3)

[When] If the above equation (3) is satisfied, the [block size of]frame is divided into blocks equal to the medium blocks BLR1 and [BLR2] BLR2, having a length [TBL/2 is selected. On the other hand, when otherwise,] TBL/2. Otherwise, i.e., the following equation (4) holds,

[MXR2/MXR1 < 10] [. . . (4)

the] and the frame remains undivided, with a block size of the large block BL having a length [TBL is selected.] TBL.

MXR2/MXR1 < 10 . . . (4)

[Here, Fig. 9 shows the procedure in realizing by software the processing from data input of respective words of an input digital signal to the] Fig. 9 shows a software routine for processing each frequency range signal prior to orthogonal transform processing. Each frequency range signal comprises plural words. In Fig. 9, at step S111, the absolute [values of respective words are] value of each word is first calculated. At the next step S112, [a] the maximum absolute value is detected. [In place] Instead of detecting the maximum absolute value, a logical sum operation may be

performed. At the next step S113, [detection of a] it is determined whether the maximum absolute value of all the words in [one block or discrimination as to whether or not logical sum operation thereof is completed is carried out. This block is a block of respective selectable block sizes. When it is discriminated] the sub block or whether the logical sum of all the words in the sub block has been taken. The sub block is an integral fraction of the frame, for instance one half (first example above) or one fourth (second example above) of the frame. When it is determined at the step S113 that the logical sum operation (or the absolute maximum value determination) of all the words is not completed (NO), the operation returns to the step S111. [In contrast,] On the other hand, when the logical sum operation (or absolute maximum determination) of all the words is completed (YES), the operation proceeds to the next step S114.

[Here, in the case of taking a] At step S114, if the logical sum of the absolute values in [a] the sub block [at the]is taken at step S112, [the] processing [for detecting a] to detect the maximum absolute value in [a] the sub block [becomes] is unnecessary. [Thus, floating] Floating coefficients (shift quantities) can be determined by [a] simple processing including only a logical sum operation.

The steps S114 and S115 [correspond to] provide the operation for [detecting a] determining the shift quantity as [a]the block floating coefficient. At the step [5114,] S114, a left shift is carried out. At the step [5115,] S115, it is determined whether [or not the fact that a] the Most Significant Bit (MSB) of the shift result is equal to ["1" is detected is discriminated.] [When] "1." If a "1" is not detected as the MSB at the step S115 (NO), the operation returns to the step [5114. In contrast, when]S114. Otherwise, if a "1" is detected (YES), the operation proceeds to the next step S116.

At the step S116, it is determined whether [or not a] the maximum absolute value (or shift quantity) of all sub blocks of the [respective sizes is obtained is discriminated.] [When the discriminated] different sizes has been obtained. When the result is NO, the operation returns to the step S111. [In

contrast, when the discriminated] Otherwise, if the result is YES, the operation proceeds to the next step S117. At the step S117, [a] the block size is determined [on the basis of] using the above equation (1) or the above equations (2) to [(4) to calculate a](4), and the maximum absolute value (or logical sum) of the block thus [determined.] determined is calculated. At the next step S119, the words in the determined block are normalized [is discriminated.] (i.e., are [At the next step S119, respective words are normalized (are] subjected to floating processing). At step [5120,] S120, it is determined whether [or not] all the words in the determined block have been normalized. [are normalized is discriminated. When the discriminated] If the result is NO, the operation returns to the step S119. [In contrast, when the discriminated] Otherwise, if the result is YES, the operation proceeds to the next step S121. At the step S121, [discrimination is made as to whether or not,] it is determined whether, when, e.g., the block size of the medium blocks BLR1 and BLR2 or the small blocks BLS1, BLS2, BLS3 and BLS4, [etc] is selected, the processing with respect to all blocks in the [range of the large block BL is] frame has been completed. [When the discriminated] If the result is NO, the operation returns to the step S111. [In contrast, when the discriminated] Otherwise, if the result is YES, the operation proceeds to the next step S122. At the step S122, the orthogonal transform processing is carried out. The processing is thus completed. In accordance with this embodiment, by [commonly] using the maximum absolute [values] value (or logical [sum outputs)] sum) calculated [every respective blocks in determination of] for each block to determine both the block floating coefficient and the block size, [a] the quantity subject to processing can be reduced. Thus, the number of steps, when, e.g., [in the case of carrying out a] processing [by using the so-called microprogram] is carried out using a microprogram, can be reduced.

[FIG.] Fig. 10 is a circuit diagram showing, in [a] block form, the outline of the configuration of an actual example of the [allowed] allowable noise calculation circuit 20. In [FIG.] Fig. 10, input terminal 21 is supplied

with [spectrum data on the frequency base] spectral coefficients from the respective DCT circuits 13, 14 and 15. [As this data, an amplitude value of the] An amplitude value and a phase value are calculated [on the basis of a] from the real number component and [an] the imaginary number component of [DCT coefficient data obtained as the result of execution of DCT operation is used.] each spectral coefficient. This approach is employed in consideration of the fact that the [hearing sense of the human being is generally sensitive for the amplitude (level, intensity) on the frequency base, but is considerably dull for the phase.] human sense of hearing is considerably more sensitive in the frequency domain to amplitude than to phase.

[Input data on the frequency base is sent to an]The resulting amplitude values in the frequency domain are sent to the energy calculation circuit 22 [every frequency band, at] in which an energy [every] for each critical band is determined [by using,] by, e.g., [a method of] calculating the sum total of the respective amplitude values in [a corresponding] the critical band, or any other appropriate method. [In place of an] Instead of determining the energy [every] in each critical band, there are instances where a peak value, or a mean value of the amplitude [value] values in the band may be used. [An] The output from the energy calculation circuit 22, e.g., a spectrum of [sum total value of] the energy sum in each respective [bands]critical band is generally called a bark spectrum. [FIG.] Fig. 11 shows such a bark spectrum SB [every respective critical bands.] [It is to be noted that, in order to simplify illustration in Fig. 11, the number of bands of the critical bands is represented by]for each critical band. To simplify the figure, only twelve bands (B1 to [B12).] B12) are shown.

[Here, in order to] To allow for the influence [in the so-called] of the masking of the bark spectrum SB, [such a]convolution processing is implemented to multiply the bark spectrum SB [to multiply it by a] by predetermined [weighting function] filter coefficients and to add the multiplied results. To realize this, [an] the output from the energy calculation circuit 22

[every] in each critical band, i.e., respective values of the bark spectrum [SB are] SB, is sent to [a] the convolution filter circuit 23. This convolution filter circuit 23 comprises, e.g., [a plurality of] plural delay elements for sequentially delaying input data, [a plurality of] plural multipliers (e.g., 25 [multipliers corresponding to respective bands)] multipliers, one for each critical band) for multiplying the outputs from [these] the delay elements by filter [coefficients (weighting function),] coefficients, and a sum total adder for [taking a sum total of respective] summing the multiplier outputs. By this convolution processing, the sum total of the portion indicated by dotted lines in [FIG. 11 is taken. It is to be noted that the above-mentioned masking refers to the phenomenon that a signal becomes inaudible as the result of the fact] Fig. 11 is calculated.

Masking is a psychoacoustic phenomenon in which a signal is rendered inaudible if it is masked by another signal. [As the masking effect, there are the time base masking effect by an audio signal on the time base and the simultaneous masking effect by an audio signal on the frequency base. Namely, by this masking effect, even if there is any noise at the portion] There is temporal masking, in which a signal is masked by a signal occurring before or after it in time. There is also simultaneous masking, in which a signal is masked by a simultaneously-occurring signal of a different frequency. As a result of masking, if there is any noise in a portion of the spectrum subject to masking, such [a] noise will be inaudible. For this reason, [in]with an actual audio signal, any noise within [a range subject to masking is considered as an] the masking range of the signal is inaudible, and is regarded as allowable noise.

[Attention is now drawn to an] An actual example of [multiplication] the filter coefficients [(filter coefficients) of] of the respective multipliers of the convolution filter circuit [23.] 23 will now be described. Assuming that the coefficient of a multiplier M corresponding to an arbitrary band is 1, the multiplying operation is carried out as follows: at the [multiplier Mt, the filter coefficient 0.15 is multiplied by outputs from respective delay elements; at the multiplier M-2, the filter coefficient 0.0019 is multiplied by those outputs; at

the multiplier M-3; the filter coefficient 0.0000086 is multiplied by those outputs; at the multiplier M+1, the filter coefficient 0.4 is multiplied by those outputs; at the multiplier M+2, the filter coefficient 0.06 is multiplied by those outputs; and at the multiplier M+3, the filter coefficient 0.007 is multiplied by those outputs.] multipliers M-1, M-2, M-3, M+1, M+2, and M+3, the outputs from the respective delay elements are multiplied by the filter coefficients of 0.15, 0.0019, 0.0000086, 0.4, 0.06, and 0.007, respectively. Thus, convolution processing of the bark spectrum SB is carried out. [It is to be noted that] M is an arbitrary integer of 1 to 25.

[Thereafter, an] The output of the convolution filter circuit 23 is sent to a [subtracter 24.] subtractor 24. [This subtracter 24 serves to determine a level a corresponding to an allowable noise level which will be described later in the convoluted region. It is to be noted that] The subtractor 24 determines the level [a] corresponding to the allowable noise level [(allowed noise level) is such a level to become in correspondence with the allowed noise level every band of the critical band by carrying out deconvolution processing as described later.] in the convoluted region. [This subtracter 24 serves to determine a level a corresponding to] The level is the level that gives an allowable noise level [which] for each critical band by deconvolution as will be described [later] below. An allowed function (i.e., a function [Here, an allowed function (function] representing the masking level) for determining the level [a] is delivered to the [subtracter] subtractor 24. By increasing or decreasing this allowed function, control of the level [a] is carried out. [This] The allowed function is delivered from [a (n-ai)] the (n - ai) function generator [25] 25, which will be described later.

[Namely, when the number given in order from a lower frequency band of bands of the critical band is assumed to be i, the] The level [a] of corresponding to the [allowed] allowable noise level is determined by the following equation:

= S - [(n-ai)]   [. . .]   [(5)] (n - ai)

(5)

[where n and a are respectively constants (a>0), and S is] where i is the number of the critical band, 1 being the number of the lowest frequency critical band, n and a are constants, a is greater than 0, S is the intensity of [a] the convolution processed bark [spectrum. In the above equation (1), (n-ai) represents an] spectrum, and (n - ai) is the allowed function. In this embodiment, n is set to 38 and a is set to 1. [There] This provides satisfactory results with no degradation of sound [quality at this time. Satisfactory coding is thus carried out.] quality.

In this way, the level [a is determined.] [This data] is determined, and is transmitted to [a divider 26.] [This divider 26 serves to apply] the divider 26, which applies deconvolution to the level in the convoluted region. Accordingly, by carrying out this deconvolution, a masking spectrum is provided from the level . [Namely, this] This masking spectrum becomes [an allowed] the primary allowable noise spectrum. It is to be noted [that] that, while [the above-mentioned] normally deconvolution processing requires a complicated operation, [simplified] a simple divider 26 is used in this embodiment to carry out deconvolution.

Then, the [above-mentioned] masking spectrum is transmitted to [a subtracter] the subtractor 28 through [a] the synthesis circuit 27. Here, the [subtracter] subtractor 28 is supplied with [an] the output of the energy [detector 22 every]calculating circuit 22 for each critical band, i.e., the [previously described] previously-described bark spectrum [SB] SB, through [a] the delay circuit 29. Accordingly, at [this subtracter 28, a subtractive] the subtractor 28, a subtraction operation between the masking spectrum and the bark spectrum SB is carried out. Thus, as shown in [FIG.] Fig. 12, the portion of the bark spectrum [S8 of which] SB having a level [is] lower than [the level indicated by] the level of the masking spectrum MS is subjected to masking.

[An] The output from the [subtracter] subtractor 28 is taken out through [an allowed] the allowable noise corrector 30 and the output terminal 31, and is

sent to a [R0M,] ROM, etc. (not shown) [at] in which, e.g., allocated bit number information are stored. [This R0M, etc. serves to output allocated] The ROM, etc. provides quantizing bit number information [every band in dependency upon an output (difference level between energy of each band and an output of the noise level setting means)] for each critical band in response to the output obtained through the [allowed] allowable noise corrector 30 from the [subtracter 28.] [This allocated] subtractor 28 (i.e., in response to the level difference between energy in each critical band and the output of the allowable noise calculating circuit). The quantizing bit number information is sent to the adaptive bit allocation [encoder 18, whereby spectrum data on the frequency base] circuit 18, (Fig. 1) where the spectral coefficients from the DCT circuits 13, 14 and 15 are quantized [by bit] using numbers of bits allocated [every respective bands.] to each critical band.

[Namely, in short, the] The adaptive bit allocation [encoder 18 serves to quantize spectrum data every respective bands by bit numbers allocated in dependency upon levels of differences between energies of respective bands of the critical band and an output of the noise level setting means. It is to be noted that a] circuit 18 quantizes the spectral coefficients in each band using the number of bits allocated in response to the difference in energy between respective critical bands and the output of the allowable noise calculating circuit. The delay circuit 29 is provided [in order] to delay the bark spectrum SB from the energy [detector 22 by taking into consideration delay quantities at respective] calculation circuit 22 to take account of delays in the circuits preceding [to] the synthesis circuit 27.

[Meanwhile, in synthesis at the above-described]The synthesis circuit [27, it is possible to synthesize data indicating the so-called minimum audible curve RC which is the hearing sense] 27 synthesizes the minimum audible level curve RC and the masking spectrum MS. The minimum audible level curve is a characteristic of the human [being as shown in FIG. 13 delivered from a minimum audible curve generator 32 and the above-mentioned masking

spectrum MS. In this minimum audible curve, if the noise absolute level is] sense of hearing, as shown in Fig. 13, and is delivered from the minimum audible level curve generator 32. According to the minimum audible level curve, noise having an absolute level below the minimum audible [curve, this noise] level curve cannot be heard. [Furthermore, even if coding is the same,]Even with the same coding, the minimum audible level curve [would vary, e.g., in dependency upon variation of a reproducing] depends on the volume at the time of reproduction. However, [it is to be noted that,] since there is not [so] a great variation in the manner in which a music [enters,] is represented by, e.g., [16 bit] the 16-bit dynamic range in actual digital systems, if it is assumed that [quantization] the quantizing noise [of, e.g.,] in the frequency band in which the ear is most [easily heard to ear]sensitive, i.e., in the vicinity of 4 [KHz, quantization] kHz, quantizing noise less than the level of the minimum audible [curve is considered to be not heard] level curve can be regarded as being inaudible in other frequency bands. [Accordingly, when a way of use in which noise, e.g., in the vicinity of 4 KHz of a word length that the system has is not heard is assumed to be employed, and an allowed noise level is provided] Therefore, if it is assumed that the system is used such that the quantizing noise near 4 kHz, for a certain quantizing word length, is inaudible, and that the allowable noise level is obtained by synthesizing the minimum audible level curve RC and the masking spectrum MS, [the allowed noise level in this case is permitted to be the level up to the portion indicated by slanting lines in FIG. 13. It is to be noted that, in this] then the allowable noise level in each critical band will be the greater of the level of the minimum audible level curve and the masking level. This is shown by the hatched lines in Fig. 13. In the present embodiment, the level of [4 KHz of] the minimum audible [curve is caused to be in correspondence with] level curve at 4 kHz is matched to the minimum level corresponding to, e.g., quantizing using 20 bits. [In FIG. 13,] Fig. 13 also shows the signal spectrum [SS is shown together.] SS.

[It is to be noted that, as] As explained above with reference to Figs. 3 to 6, in the critical [band] bands where the minimum audible [curve is considered as an allowed noise,]level is selected as the allowable noise level, quantizing bit allocation [every small bands obtained] is performed by dividing the critical band into [smaller bands is carried out.] [Namely, at the comparator 35, the] sub bands. The minimum audible level curve from the minimum audible level curve generator 32 and the masking spectrum MS from the divider 28 are compared with each [other. The compared] other in the comparator 35. The result is sent to the synthesis circuit [27,] 27. [out as a] The flag FRC [and] is taken out [as a flag FRC] from output terminal 36. For example, in the bands B11 and B12 of Fig. [13.] 13, since the level of the minimum audible level curve RC is higher than the level of the masking spectrum MS, [this] the minimum audible [curve RC is regarded as an allowed noise,] level curve level RC is selected as the allowable noise level, and the flag FRC is set to 1. Thus, the level of the minimum audible [curve RC, e.g., on the lowest frequency side when the critical band is finely divided will be transmitted. As previously described above, calculation of allowed noise levels every respective bands is carried out on the decoder side.] level curve RC for the lowest frequency of the sub bands into which the critical band is divided will be transmitted. As described above, calculation of allowable noise levels for the other sub bands is carried out in the expander.

[At the allowed noise level corrector 30, the allowed noise level in an output from the subtracter 28 is corrected on the basis of information of, e.g., equi-loudness curve sent from a correction information output circuit 33. Here, the equi- loudness curve is a characteristic curve relating to the hearing sense characteristic of the human being, and is a curve formed by determining sound pressures of sound at respective frequencies which is heard at the same intensity of that of a pure sound, e.g., at 1 KHz to connect them. This equi-loudness curve is also called an equi-sensitivity curve of loudness.] [Further, this equi-loudness curve is substantially the same curve as the minimum audible

curve RC shown in FIG. 13. In the equi-loudness curve, for example, in the vicinity of 4 KHz, sound is heard at the same intensity as that at 1 KHz even if the sound pressure is lowered by B to 10 dB as compared to that at 1 KHz. In contrast, in the vicinity of 50 KHz, sound cannot be heard at the same intensity as the at 1 KHz unless the sound pressure at 50 KHz is higher by about 15 dB than that at 1 KHz. For this reason, it is seen to allow a noise (allowed noise level) above the level of the minimum audible curve to have a frequency characteristic given by a curve corresponding to the equi-loudness curve. From these facts, it is seen that it is adapted for the hearing sense characteristic of the human being to correct the allowed noise level by taking the above-mentioned equi-loudness curve into consideration.]

The allowable noise level corrector 30 corrects the allowable noise level at the output of the subtractor 28 in response to information regarding, e.g., the equal loudness curve from the correction information output circuit 33. The equal loudness curve is a curve characterizing another characteristic of the human sense of hearing. The equal loudness curve corrects sound pressure levels at different frequencies so that they are perceived as sounding as loud as a pure sound at 1 kHz. The equal loudness curve has substantially the same characteristic as the minimum audible level curve RC shown in Fig. 13.

According to the equal loudness curve, a sound in the vicinity of 4 kHz is perceived as being as loud as a sound at 1 kHz having a sound pressure level 8 to 10 dB higher. On the other hand, a sound in the vicinity of 50 Hz must have a sound pressure level some 15 dB higher than a sound at 1 kHz sound to be perceived as sounding as loud. Because of this, the allowable noise level must be corrected using the equal loudness curve to adjust the allowable noise level for the loudness sensitivity of the human sense of hearing.

[Here, the correction information output circuit 33 may be of such a structure to correct the above- mentioned allowed noise level on the basis of information of an error between a detected output of output information quantity (data quantity) in quantization at the encoder 18 and a bit rate target

value of the final coded data.] Additionally, the correction information output circuit 33 may also correct the allowable noise level in response to the difference between the actual number of bits used by the adaptive bit allocation circuit 18 (Fig. 1) to quantize the spectral coefficients, and the target number of bits, which is the total number of bits available for quantizing. The reason why such a correction is made is as follows. [In general, there are instances where total number of bits obtained by applying, in advance, temporary adaptive bit allocation to all bit allocation unit blocks may have an error with respect to a fixed bit number (target value) determined by a bit rate of the final coded output data. In such instances,] There are instances in which there is an error occurs between the total number of bits allocated by the primary bit allocation process and the target number of bits, which is determined by the bit rate of the compressed digital signal. In such instances, the quantizing bit allocation is made for a second time [so that the above-mentioned error becomes equal to zero. Namely, an approach is employed such that when the total allocated bit number] to reduce the error to zero. For example, if the total number of bits allocated is less than the target value, [bit numbers of difference are allocated to respective unit blocks to add insufficient bits, while when the total allocated bit number is greater than the target value, bit numbers of difference are allocated to respective unit blocks to reduce surplus bits.] a number of bits equal to the difference between the actual number of bits and the target number of bits is allocated among the critical bands to provide additional bits. Alternatively, if the actual number of bits is more than the target number of bits, a number of bits corresponding to the difference between the actual number of bits and the target number of bits is removed from the critical bands to remove excess bits.

[To carry out this, an error from the target value of the total allocation bit number is detected.] [In dependency upon this error data, the] To correct the actual number of bits, the difference between the actual number of bits and the target number of bits is measured and the output correction information output circuit 33 [outputs] provides correction data [for correcting respective

allocated bit numbers.] [Here, in the case where the above-mentioned] that is used to correct the numbers of bits allocated to the critical bands. Where the error data indicates that insufficient [bit number,] bits have been allocated, an increased number of bits are used per [unit block. Thus, consideration can be made in connection with the case where the data quantity is greater than the target value. In contrast, in the case where the above- mentioned error data is data indicating remainder of bit number, a lesser number of bits can be used per each unit block. Thus, consideration can be made in connection with the case the data quantity is less than the target value. Accordingly, from the] critical band. Conversely, where the error data indicates that excess bits have been allocated, fewer bits can be used in each critical band. The correction information output circuit [33, in dependency upon this error data,] 33 provides data [of] for the correction value for correcting [an allowed] the allowable noise level [in an] at the output from the [subtracter] subtractor 28, e.g., on the basis of information data of the [equi-loudness curve is outputted. A]equal loudness curve, in response to the error data. The correction value [as described above] is transmitted to the [allowed]allowable noise [corrector] level correction circuit 30. Thus, the [allowed] allowable noise level from the [subtracter] subtractor 28 is corrected.

[Further, there may be employed a configuration such that the] The above-described synthesis processing for the minimum audible level curve [is not carried out.] may be omitted. In this case, minimum audible level curve generator 32 and synthesis circuit 27 [become] are unnecessary, and [an] the output from the [subtracter] subtractor 24 is subjected to deconvolution at the divider 26, and is transmitted immediately to the [subtracter 28.] subtractor 28.

[Further, as shown in Fig. 14, in the case of carrying out the block] Block floating processing and [the] block floating release processing [before and after the Inverse Orthogonal Transform (IDCT, i.e., Inverse Discrete Cosine Transform) processing on a decoder side, an approach may be employed to take a logical sum of absolute values of respective words in a block, thereby making

it possible to determine a floating coefficient.] may also be applied in the expander, before and after, respectively, the inverse orthogonal transform (IDCT) processing. In the expander, the logical sum of the absolute values of the spectral coefficients for each block may be taken to determine the block floating coefficient for the block.

[In Fig. 14,] In Fig. 14, the input terminal 51 is supplied with [coded data on the frequency base as] quantized spectral coefficients obtained from the output terminal 19 of the compressor shown in Fig. 1. [This coded data] The quantized spectral coefficients is sent to [an] the adaptive bit allocation decoder 52, [at which it is subjected to decoding processing. Such data on the frequency base which have undergone adaptive bit allocation decoding processing are sent to a] where the adaptive bit allocation applied by the adaptive bit allocation circuit in the compressor is reversed. The resulting spectral coefficients are sent to the block floating processing circuit 56, [at which] where block floating processing [every block is implemented thereto. Thereafter, the data thus processed respectively undergo, at Inverse Orthogonal Transform]is applied to each block of spectral coefficients in each frequency range. Then, the blocks of block floating processed spectral coefficients are subject to inverse orthogonal transform (IDCT, i.e. Inverse Discrete Cosine Transform circuits 53, 54 and 55 in the example of Fig. 14) [processing opposite to the processing at] processing, inverse to the othogonal transform processing applied by the respective orthogonal transform circuits 13, 14 and 15 of Fig. 1. [These] The outputs from the inverse orthogonal transform circuits 53, 54 and 55 are sent to [a]the block floating release circuit 57, [at which] where block floating release processing [every block is carried out on the basis of] is applied to each block using the block floating information from the block floating processing circuit 56. [Outputs of] The resulting frequency range signals in the respective [bands] frequency ranges from the block floating release circuit 57 undergo, by using [synthetic] the synthesis filters 58 and 59, processing opposite to the processing by the band division filters 11 and 12 of Fig. [1] 1, so that [outputs in] the

respective [bands are synthesized. The output] frequency range signals are synthesized to provide a single digital output signal. The digital output signal thus synthesized is taken out from the output terminal 60.

It is to be noted that this invention is not limited to the above-described embodiment, but is applicable, e.g., not only to a signal processing apparatus for an audio signal but also to a signal processing apparatus for a digital speech signal or a digital video signal, etc.

As described above, the [coding] apparatus for a compressing a digital input signal [of] according to this invention is adapted to carry out [a] block floating processing of [an] the [input] digital input signal [by a] in variable length [block] blocks, and thereafter to implement [an] orthogonal transform processing thereto. In [this coding apparatus,] the compressor, by determining the [length of a variable length] size of each block and [a] the block floating coefficient of the block floating [on the basis of] processing in response to the same index, it is possible to reduce a quantity subject to [quantization] processing, or the number of steps of a program.

Further, in accordance with the [coding apparatus for digital signal, when allowed noise levels every critical bands are determined by] apparatus for compressing a digital input signal, when the allowable noise level for each critical band is determined using the minimum audible level, bit allocation is carried out [by allowed noise levels every small] according to the allowable noise level in sub bands obtained by further dividing [the] a critical [band to] band, and only [transmit] a flag indicating [this, thus to avoid] this is transmitted. This avoids the necessity of sending [allowed noise levels every small bands.] an allowable noise level for each sub band. Accordingly, accurate [allowed] allowable noise levels can be provided without increasing the quantity of auxiliary information [quantity.] transmitted. This [leads to the fact that] way, signal quantity can be improved without degrading [bit] the data compression efficiency. In addition, even if [an] the absolute value of the minimum audible [limit] level is altered later, compatibility can be maintained.